IN THE SPECIFICATION:

Please amend the paragraph beginning at page 1, line 6, as follows.

The present invention is related to United States Patent Application Number 09/668,243 entitled "Radio Link Protocol (RLP)/Point-to-Point Protocol (PPP) Design for Wireless Multimedia Packet Networks that Passes Corrupted Data and Error Location Information Among OSI Layers," ~~(Attorney Docket Number Lu 7-1),~~ filed contemporaneously herewith, assigned to the assignee of the present invention and incorporated by reference herein.

Please amend the paragraph beginning at page 1, line 19, as follows.

It is inevitable that future wireless services will support Internet Protocol (IP)-based multimedia applications. For example, current and emerging wireless networks allow (i) a user to download information from the Internet using a wireless communication device, (ii) Internet-to-mobile or mobile-to-mobile videoconferences, (iii) streaming of video or audio information (or both) from the Internet to a wireless communication device, and (iv) electronic-commerce applications. In general, the multimedia services can be classified into two categories. Real-time services generally have delay constraints but can tolerate channel errors, and include interactive services, such as voice, voice-over-IP, packet video/audio, videoconference applications. Non-real-time services, on the other hand, are generally sensitive to channel errors but have more relaxed latency requirements, and include Web browsing, electronic mail and file transfer protocol (FTP) applications. It is noted, however, that non-real-time services for wireless systems should still provide a reasonable level of latency in order to be compatible with performance on a wired network, such as the Internet.

Please amend the paragraph beginning at page 5, line 17, as follows.

FIG. 3 illustrates a general wireless protocol stack 310 and packet structure 320. The link layer 330 partitions each single data packet into multiple units to accomplish physical layer transmission. The unit size depends on the radio link protocol (RLP), medium access control (MAC) and physical layer (PHY) 335, but is usually much smaller than the packet size. In 3G wireless systems, for example, each physical layer frame corresponds to a transmission unit, assuming low and medium data rates. To support high data rate services, the MAC layer specifies that the RLP layer can subdivide each physical layer frame into smaller logical frames, referred to as LTUs, each

associated with a 16 ~~bits~~ bit CRC. Typical LTU size ranges from 300-600 bits (40-80 bytes), while IP packets are typically 600-1500 bytes long.

Please amend the paragraph beginning at page 5, line 26, as follows.

At the receiver 140, the RLP can specify a limited number of retransmissions to compensate for LTU losses. The RLP forwards the received LTUs to the interface, e.g., Point-to-Point Protocol ~~(PPP)~~, to reconstruct the packet. For non-real-time services, PPP forwards the packets to the TCP layer, where the packet losses after RLP retransmissions can be recovered at the TCP level through packet level retransmission and congestion control. For real-time services employing UDP, upon receiving a packet from the PPP protocol layer, the conventional UDP layer performs a packet level cyclic redundancy check(CRC) to validate the information within the packet, including both the packet header and the payload. In this case, any LTU loss would result in the whole packet being discarded. Mathematically, the packet loss rate (PER) can be approximated as:

$$PER = 1-(1-p)^m \approx mp \text{ (for large } m \text{ and small } p),$$

where m is the number of LTUs per packet and p is the LTU error rate (LER) after retransmission. Therefore, using a conventional UDP in a wireless network will yield a considerable amount of packet loss, and as a result, poor video/audio quality and increased power consumption. The inefficiency of UDP in wireless networks arises from the discarding of a packet containing only a small part of corrupted data. As such, the UDP also throws out error-free data within the packet. Indeed, applications can utilize error-free data to recover corrupted data.

Please amend the paragraph beginning at page 7, line 19, as follows.

As previously indicated, the UDP and UDP Lite protocols do not perform well with packet based FEC. One significant reason is that the UDP and UDP Lite protocols ignore useful channel information from the RLP layer 330. The present invention recognizes that such information can be exploited to maximize FEC coding efficiency. However, the current protocol design does not support information communications from the RLP layer 330 to the PPP/IP/UDP layers and above. The present invention proposes a new system protocol design that allows the exchange of certain information in both directions among the layers 310. For a more detailed discussion of communications between the RLP and PPP layers in the conventional open system interconnection (OSI) model, see co-pending U.S. Patent Application Number 09/668,243 entitled

"Radio Link Protocol ~~(RLP)~~/Point-to-Point Protocol ~~(PPP)~~ Design for Wireless Multimedia Packet Networks that Passes Corrupted Data and Error Location Information Among OSI Layers," ~~(Attorney Docket Number Lu 7-1),~~ incorporated by reference above. A revised UDP protocol, referred to herein as the complete User Datagram Protocol (CUDP), is disclosed that reduces or avoids the discarding of unnecessary packets by passing the LTU error information as well as the corrupted packets to the Packet FEC decoder. Depending on the implementation of the FEC decoder, the error information can be represented in two forms:

Please amend the paragraph beginning at page 9, line 6, as follows.

The FEC encoder 400 selects *k* packets of length X units. The proposed system encodes multiple packets of the same size together. For real-time applications, the packets should have the same or similar delay constraint, e.g., packets correspond to a single video frame. The packets can be of different length. If so, they are bit stuffed to match the longest packet length. Indeed, the source coding and packetization scheme can be designed to generate packets with equal or similar size, as described in co-pending <u>United States Patent Application</u> ~~patent application~~ <u>Number 09/668,243</u> entitled "Radio Link Protocol ~~(RLP)~~/Point-to-Point Protocol ~~(PPP)~~ Design for Wireless Multimedia Packet Networks that Passes Corrupted Data and Error Location Information Among OSI Layers," (Attorney Docket Number Lu 7-1), incorporated by reference above. The channel encoder at the application layer takes one data unit from each packet and generates $(n - k)$ parity units to construct $(n - k)$ additional packets in a vertical packet coding (VPC) structure, shown in FIG. 5A.